TAKING THE MYSTERY OUT OF SINGLE SIDEBAND

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In the previous two issues of *Ham Radio Horizons* I talked about single-sideband in general, covering the reasons for the improvement in communications ability, and about receiving ssb. Much of the information covered is still applicable to this part of the ssb picture — transmitting. The basic process of generating the sideband signal has been explained, which makes this "transmitting" portion of the discussion a bit easier. All that remains is to delve just a bit more into the generation process, and then apply the principle to developing a useful signal for some of the popular amateur bands.

Audio processing

Throughout most of this discussion I have talked about a single audio tone instead of a "band" of tones or frequencies, just to make the steps a bit easier to follow. Where reference was made to voice frequencies, the example used was limited to frequencies between 100 and 5000 hertz. I am sure that you realize that many voices cover more than that range, depending upon the gender and individual characteristics (**Fig. 1**). Actually, the range of voice frequencies used in ssb communication is made even more narrow — 300 to 3000 hertz in some instances. This practice did not apply only to the development of ssb, though. As far back as the middle 1930s the more popular ham bands were becoming crowded enough that many authorities recommended that the speech-amplifiers in a-m (amplitude modulated) transmitters be modified to limit the range of frequencies that would be amplified especially the higher tones. Not too many amateurs were enthused about giving up their "wide-range" sound on the air, but there was a small group that did follow the suggestion. Part of the great space-saving ability of ssb is that one sideband and the carrier has been eliminated, as we have seen. Almost as important is the restricted range of frequencies that the audio amplifiers and filters will allow to pass (**Fig. 1**). Considerable experience was required before a majority of amateurs were convinced that it really was possible to transmit reasonable sounding audio — good enough for communications purposes — by using only part of the voice frequencies. This not only had the benefit of decreasing the spectrum taken up by each signal, but the additional filtering involved in the receivers allowed the amateur to "hear better" as well. Transmitters could "punch through" when the conditions got rough, and the receivers could "dig down deep" in the layers of signals that sometimes appeared when a rare station made its presence known.

A second type of speech processing is very much talked about today, and it has both good and bad points. This system is one of trying to fill in the valleys between voice peaks, thereby making the transmitter more efficient. The human voice is a very complex thing, and there still is much research being done to analyze the purpose and effects of each part of the spoken sound. In addition to the wide range of tones found in the voice, there are also great variations in the strength, or amplitude, of those tones. An example of this is shown in **Fig. 2**. The greatest amplitude represented in the drawing is called the "peak." It is that part of the voice that drives the transmitter to its greatest output, so you can see that if the "peak" voltage is there for only a short period of time, then the transmitter is sending out its maximum power for a correspondingly short time. If there was some way to increase either the length of time of the peak, or to build up the portions that were not as "loud," it should cause the transmitter to produce its greatest power for a longer time.

The type of speech processing that does this little trick can perform its deed either by limiting the amplitude of the strongest parts, while allowing the weaker ones to go untouched, or it can use a complex circuit that detects how strong the signal is, and by electronically changing the gain of a stage, force the output to stay at or below a desired level. In other words, the stronger you talk, the more

off it turns the amplifier!

Either the limiting or the gain-changing method must be accompanied by numerous other circuits that do such things as keep the voice from becoming badly distorted, keep the amplifiers from creating new frequencies outside the voice range, and other housekeeping duties. There are several ways of getting to the same goal, and, since the confusion factor is high enough with just the basics of ssb, I'll not go into all that in this series.

Some of the benefits of speech processing for ssb are; **a**) you get more overall efficiency from a transmitter because it is kept working hard longer, **b**) it fills the frequency (channel) you are using with more signal, thus making it easier to receive at the other end, and **c**) it can build a weaker voice up to a real "loudmouth" level if needed. Most of the disadvantages tend to show up when a speech processing system is misadjusted or improperly used. If the overall gain is set too high, background noises tend to be almost as loud as you are — believe it or not, most people do not like to hear the dog barking next door, your kids playing upstairs, and the Saturday Night Movie from the TV set, all while you are trying to say something. Furthermore, if the amount of limiting or compression is set too great (not enough difference between the peaks and valleys) the voice loses its character and people will wonder if you are in great pain. Certainly either condition is not of great value to communication.

Another fault with the use of many processors lies in the mistaken idea that there is no limit to the amount of improvement in output power that can be obtained by simply turning up the gain knob or by talking closer or louder. The limitation comes in the transmitter elements themselves. Many tubes and transistors are working close to their rated values in Amateur equipment, and it does not take much of an increase in input from the speech end of things to make them reach their limit. Beyond that limit, any further "push" from the microphone will simply cause the transmitter to "saturate," which means that it has "had it right up to here," and cannot work any harder. Under these conditions, you are most likely doing more harm than good, because the people who want to carry on a conversation on an adjacent frequency are being "splattered on."

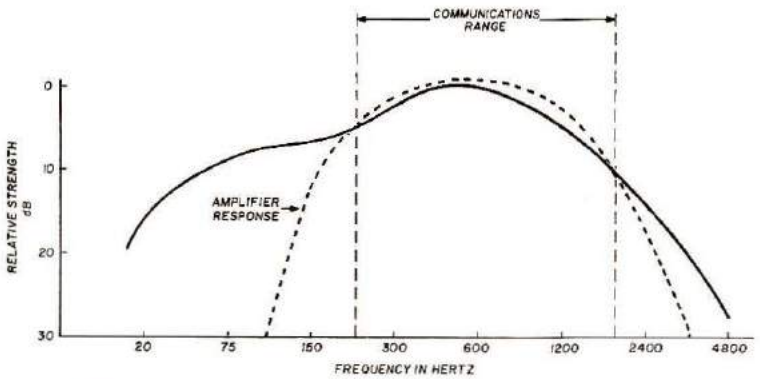

Fig. 1. The power involved in voice frequencies shows a response peak between 400 and 1000 hertz. This plot is for an average male voice when speaking at normal loudness. By selecting the proper portion of the curve, none of the essential tones necessary for communicating are lost. The dotted line shows an idealized response curve for an audio amplifier that would let the important parts of speech through. If the curve were shifted lower, the voice would sound boomy; emphasizing the higher parts would cause the voice to sound tinny and lack "punch."

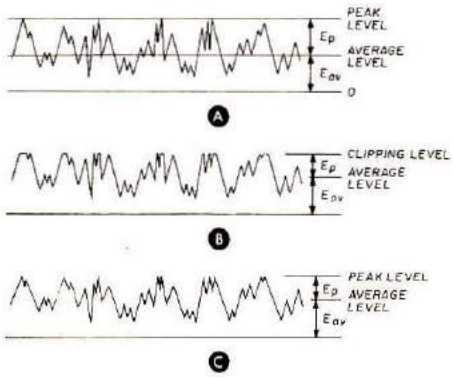

Fig. 2. Speech processors can also be designed to reshape the maximum strength of the modulating tones or voice. The variations in average voice peaks can be seen in **A**. A clipper can be designed into the circuit, and will remove the upper part of the highest peaks, **B**. A volume compressor or gain-reducing circuit will keep the original form of the peaks, but prevent them from rising as high, shown in **C**. The important thing here is that the *average* level is increased, thereby allowing better efficiency in the transmitter and improved signal readability.

Modulator or mixer

In earlier portions of this series, I used the terms "balanced modulator," "balanced mixer," and "mixer," from time to time. In many ways they are alike — practically interchangable. It's just that if you are using a given circuit as a part of a transmitter that produces a sideband signal from voice, then you call it a modulator. If you happen to use the same one for the purpose of translating a low-frequency to a higher one, or vice-versa, then it is more properly called a mixer. The balanced part stems from the application of the voltages in such a manner that the whole circuit is working, rather than having one part of the circuit working against ground (**Fig. 3**).

In part 1, I explained how a sideband was generated, and in part 2 I showed how it could be translated to a particular Amateur frequency (and then "detranslated" so you could hear it). Now I'll refresh your memory and at the same time use as a practical example some frequencies that might

be found in a piece of modern ssb gear.

In **Fig. 4**, you can see the familiar balanced modulator that has been talked about before, along with a speech amplifier and a carrier generator. Since this is not a construction article, I have not shown all of the components needed to build a circuit like this. However, there have been many very similar circuits published in Amateur literature, so if you would like to explore further, you can do so. (See References for additional reading suggestions). The carrier used here is 9 MHz, and the carrier-generator oscillator is crystal controlled. Either upper or lower sideband can be obtained by simply switching the crystal that is in use.

Following the balanced modulator you will notice a filter. This is a crystal filter that has the response curve necessary to select the right portion of the sideband and reject the unwanted parts, along with the unwanted sideband. (Refer to **Fig. 5** in the March issue). The filter stage is followed by an amplifier, necessary because the filter has an appreciable loss (you get less out than you put in), which is, in turn, followed by a mixer.

This mixer is a "translating" stage, or (here *he* comes again!) it *heterodynes* the 9-MHz sideband signal to an Amateur band. Here I have placed more crystals and a vfo in the circuit. Thus you can select an output frequency in the 3.8-, 7-, 14-, 21-, or 28-MHz band. Nothing hard about that — it is nearly the same as I told you about last month in the receiver portion of the series. But, with one of these switch positions I am pulling a sneaky on you. When you switch the vfo to obtain an output on the 3.8-MHz band, the voice comes out upside down! (Remember **Fig. 4** in the April issue?) There's an easy cure for that, however — you simply connect this set of crystals and the 9-MHz carrier generator crystals to the same switch, and wire them so that it changes the crystal at the same time you change bands. How about that! Incidently, the voice starts out upside-down, right from the modulator and filter. It has to because of the frequency translation that takes place in the second mixer for the bands 40 through 10 meters. The mixer stage should be followed by several

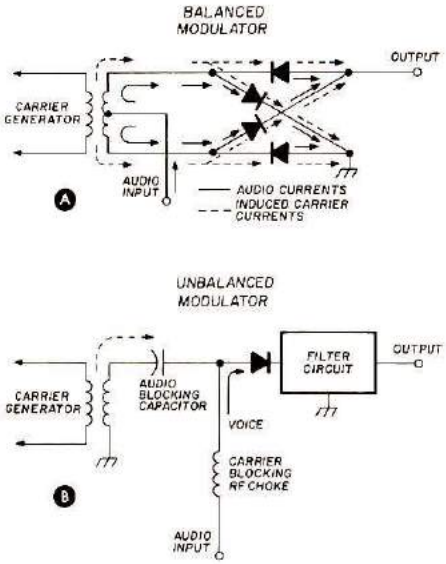

Fig. 3. The balanced modulator **A**, is used in many ssb-generating circuits because of its greater efficiency and because the carrier more effectively cancels itself after performing the basic function of turning the diodes on and off at a rapid rate. This is in contrast to the unbalanced modulator (or mixer) at **B**, which requires good filters at the output to remove leftover carrier energy.

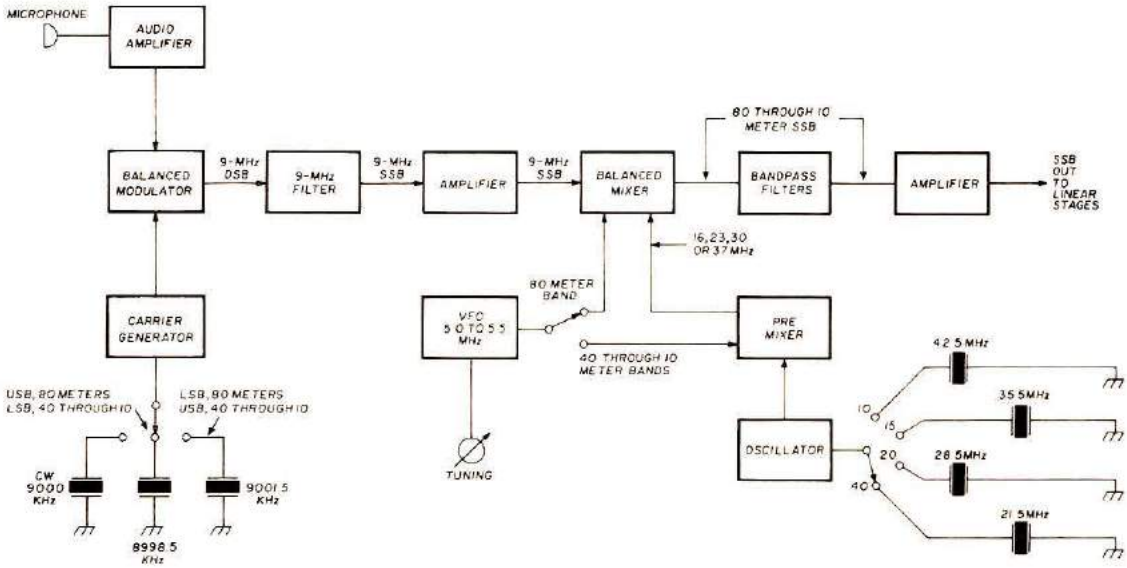

Fig. 4. A simplified block diagram of an ssb transmitter that is typical of many units in use today. The choice of frequencies to use for generating the basic ssb signal is determined somewhat by the availability of good filter components. Fortunately there are several good crystal filters available in the hf range between 3 and 21 MHz. However, great care must be used in selecting all of the frequencies involved because unwanted mixing products are often difficult to eliminate. They could cause trouble by being transmitted along with the correct product, or could block the receiver with a very strong signal.

amplifiers, of course, because the power level at this point is very low — only a few tenths of a watt. Several stages can be used, one after another, until the power is built up to whatever level is needed — hundreds or thousands of watts.

The "third method"

I have already talked about two common types of ssb generated circuits, the filter type and the phasing type. There is another way of obtaining a sideband signal; one quite popular in Europe and England. Generally, it is referred to as the "third method." It may eventually gain in popularity over a larger part of the world because it makes use of some digital circuitry, which is the "in" thing these days.

A block diagram of a system using the "third method" is shown in **Fig. 5**. The basic generation can be accomplished in six integrated circuits. Other circuitry, not shown, is required to provide the clock input, and of course there must be an audio amplifier stage and a vfo to complete the picture. In one example of this system, reported in the Radio Society of Great Britain publication *Radio Communication*, October, 1974, the designer (A. de Muijnkc of Holland) used 7400 Hz as the clock input and a vfo frequency of 14 MHz, to provide ssb output in the 3.5- to 4-MHz band. (See References for further reading).

It is basically a phasing type of generator, with the phase changes taking place in the frequency-divider ICs. For those of you who are not familiar with the workings of digital circuitry, let me point out that the output of a divider, such as the 7473 used here, consists of a pulse from what is termed the "Q" output, and another from what is called the "not-Q," represented by an overscored Q ($\overline{Q}$). The $\overline{Q}$ output is off whenever the Q is on, and vice versa, thus one IC can give you two outputs, which are 180 degrees out of phase with each other, and an output from the K section that is 90° out of phase with both the Q and the $\overline{Q}$ pulses.

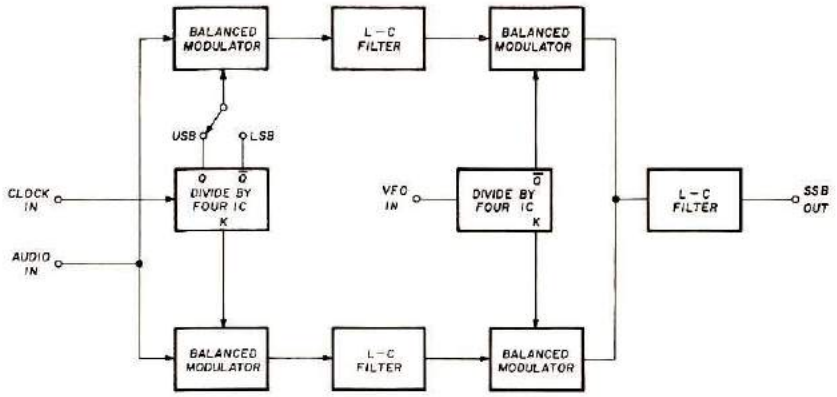

Fig. 5. A block diagram of a "third-method" of developing ssb. Although it is a variation of the phasing system, it makes use of modern components and digital techniques to an extent. Because of the upper frequency limit of most divider ICs, the system works to approximately 8 MHz. Improved integrated circuits could extend this limit. The "clock" mentioned in the text and shown here has nothing to do with timekeeping in the normal sense, but rather designates the function of turning a portion of the circuit on and off at a periodic rate.

An advantage to this method of ssb generation is that the filters needed are relatively simple and inexpensive, consisting mainly of inductance and capacitance in combination. Another plus is that the system can directly produce a variable-frequency output, without the addition of an external mixer that accepts the ssb and the vfo input. A disadvantage, at the moment, is that the frequency-divider ICs most commonly available will operate only to 30 or 40 MHz. Since the vfo input must be at four times the expected output frequency, this puts a limit on the number of bands that can be covered with present ICs. There are some dividers available that work well into the uhf range, but the expense is prohibitive.

Conclusion

This series of ssb exploration articles has been intended to help the beginner to understand some of what goes on inside the many boxes that send and receive that mode. There are many variations in the approach to building an ssb transmitter, receiver, or transceiver. However, they must all follow one of the three methods I have talked about, at least until someone finds a "fourth method." A reference section follows, containing some recommended reading for those who would like to delve deeper into the principles, circuitry, and perhaps the ways in which you can build your own.

References

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